User Manual

GL-VP-6222 и
GL-VP-6244

Analog IP Gateway

2FXS&2FXO Ports OR 4FXS&4FXO Ports

http://www.giga-link.ru

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1 WELCOME

Thank you for purchasing the Gigalink GL-VP-62XX Analog FXS IP Gateway. The GL-VP-62XX offers an easy to manage, easy to configure IP communications solution for any business with virtual and/or branch locations. The GL-VP-62XX supports popular voice codecs and is designed for full SIP compatibility and interoperability with 3rd party SIP providers, thus enabling you to fully leverage the benefits of VoIP technology, integrate a traditional phone system into a VoIP network, and efficiently manage communication costs.

This manual will help you learn how to operate and manage your GL-VP-xxxx FXS Analog IP Gateway and make the best use of its many upgraded features including simple and quick installation, multi-party conferencing, This IP Analog Gateway is very easy to manage and scalable, specifically designed to be an easy to use and affordable VoIP solution for the small – medium business or enterprise.

1.1 Gateway GL-VP-62XX Overview

The new GL-VP-62XX series has a compact and quiet design (no fans) and offers superb audio quality, rich feature functionality, strong security protection, and good manageability. It is auto-configurable, remotely manageable and scalable.

The GL-VP-62XX features 4 or 8-port FXS interface for analog telephones, dual 10M/100Mbps network ports with integrated router, PSTN life line in case of power failure,. In addition, it supports the option of 2 SIP Server profiles, caller ID for various countries/regions, T.38 fax, flexible dialing plans, security protection (SIPS/TLS), comprehensive voice codecs including G.711 (a/u-law), G.723.1, G.726(16/24/32/48 bit rates), G.729A/B/E and iLBC(Pending).

Caution: Changes or modifications to this product not expressly approved by Gigalink Co LTD, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.

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1.2 Safety Compliances

The GL-VP-62XX is compliant with various safety standards including FCC/CE. Its power adaptor is compliant with UL standard. Warning: use only the power adapter included in the GL-VP-62XX package. Using an alternative power adapter may permanently damage the unit.

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1.3 Warranty

Gigalink has a reseller agreement with our reseller customer. End users should contact the company from whom you purchased the product for replacement, repair or refund.

If you purchased the product directly from Gigalink, contact your Gigalink Sales and Service Representative for a RMA (Return Materials Authorization) number. Gigalink reserves the right to remedy warranty policy without prior notification.

2 CONFIGURE YOUR GL-VP-62XX

Connecting your GL-VP-62XX is easy. Before you begin, please verify the contents of the GL-VP-62XX package.

2.1 Equipment Packaging

Unpack and check all accessories. The GL-VP-62XX package contains:

- One GL-VP-62XX VoIP adapter
- One universal power supply
- One Ethernet cable

2.2 Connect The GL-VP-62XX

Managing the GL-VP-62XX gateway and connecting the unit to the VoIP network is very simple. Follow these four (4) steps to connect your GL-VP-62XX gateway to the Internet and access the unit's configuration pages.

1. Connect standard touch-tone analog phones to the FXS1-FXS4 ports.

2. Insert the Ethernet cable into the WAN port of GL-VP-62XX and connect the other end of the Ethernet cable to an uplink port (a router or a modem, etc.)

3. Connect a PC to the LAN port of GL-VP-62XX for initial configuration or if it is being used as a router.

4. Plug the power adapter into the GL-VP-62XX and into a power outlet.

2.3 GL-VP-62XX Back Panel

Figure 1: Diagram of GL-VP-62XX Back Panel

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TABLE 1: Definitions Of The GL-VP-xxxx Connectors

LAN	Connect the LAN port with an Ethernet cable to your PC.
WAN	Connect to the internal LAN network or router.
PSTN Line	1 port
RESET	Factory Reset button. Press for 8 seconds to reset factory default settings.
DC 9V 2A	Power adapter connection
FXS1 – FXS4	FXS port to be connected to analog phones / fax machines
FXO1 – FXO4	FXO ports to be connected to physical PSTN lines from a traditional PSTN PBX or PSTN Central Office.

Once the GL-VP-xxxx is turned on and configured, the front display panel indicates the status of the unit.

2.4 GL-VP-62XX Display Panel

Figure 2: Diagram Of GL-VP-62XX Display Panel

TABLE 2: Definitions Of The GL-VP-xxxx Display Panel

Power LED	Indicates Power. Remains ON when Power is connected and turned ON.
RUN LED	blinking after boot-up.
LAN LED	Indicates LAN port activity

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WAN LED	Indicates WAN port activity
LEDs 1 - 8	Indicate status of the respective FXS Ports on the back panel **Busy** - ON (Solid Green) **Available** - OFF Slow blinking FXS LEDs indicates Voice Mail for that port.

NOTE:

- Flast blinking of RUN, WAN LED together indicates a firmware upgrade or provisioning state.
- LEDs POWER, and WAN are ON and READY blinking when device is up and running.

3 GL-VP-62XX FEATURES

The GL-VP-62XX is a next generation IP voice gateway that is interoperable and compatible with leading IP-PBXs, SoftSwitches and SIP platforms. The GL-VP-62XX series is auto-configurable, remotely manageable and scalable. There are two models, the GL-VP-62XX, each offering superb voice quality, traditional telephony functionality, easy deployment, and 2 or 4 FXS&FXO ports respectively. Each model features flexible dialing plans, PSTN failover, integrated call routing to support a pure IP network call and an external power supply.

3.1 Software Features Overview

- 2 or 4 FXS ports
- 2 or 4 FXO ports
- Two RJ-45 ports (switched or routed)
- Multiple SIP accounts & profiles (2 or 4 accounts / choice of 2 profiles per account)
- Supports Voice Codecs: G711(a/μ, Annex I & II), G723.1A, G726 (ADPCM with 16/24/32/40 bit rates), G729 A/B/E, Ilbc(Pending)
- T.38 Fax
- Comprehensive Dial Plan support for Outgoing calls.
- G.168 Echo Cancellation
- Voice Activation Detection (VAD), Comfort Noise Generation (CNG), and Packet Loss Concealment (PLC)
- Supports PSTN/PBX analog telephone sets or analog trunks

TABLE 3: GL-VP-62XX SOFTWARE FEATURES

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	GL-VP-62XX FXS Analog Gateway Series
Telephone Interfaces	GL-VP-xxxx 6104: 4 ports, 4 SIP accounts & choice of 2 profiles GL-VP-xxxx 6108: 8 ports, 8 SIP accounts & choice of 2 profiles FXS/FXO, RJ-11
Network Interface	Two (2) 10M/100 Mbps, RJ-45
LED Indicators	Power and Line LEDs
Voice over Packet Capabilities	Voice Activity Detection (VAD) with CNG (comfort noise generation) and PLC (packet loss concealment), AEC with NLP, Packetized Voice Protocol Unit (supports RTP/RTCP and AAL2 protocol), G.168 compliant Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711
Voice Compression	G.711 + Annex I (PLC), Annex II (VAD/CNG format) encoder and decoder, G.723.1A, G.726(ADPCM with 16/24/32/40 bit rates), G.729A/B/E, iLBC G.726 provides proprietary VAD, CNG, and signal power estimation Voice Play Out unit (reordering, fixed and adaptive jitter buffer, clock synchronization), AGC (automatic gain control), Status output, Decoder controlling via voice packet header
DHCP Server/Client	Yes, NAT Router or Switched Mode
Fax over IP	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through, Fax Datapump V.17, V.19, V.27ter, V.29 for T.38 fax relay
QoS	Diffserve, TOS, 802.1 P/Q VLAN tagging
IP Transport	RTP/RTCP
DTMF Method	flexible DTMF transmission method, User interface of In-audio, RFC2833, and/or SIP Info
IP Signaling	SIP (RFC 3261)
Provisioning	TFTP, HTTP, HTTPS (pending)
Control	TLS/SIPS
Management	Syslog support, HTTPS (pending), Telnet, remote management using Web browser
Dial Plan	Yes
UPnP Support	Yes
Power	Output: 9VDC / Input: 100–240 VAC/50-60 Hz
Mounting	Rack mount, Wall mount, Desktop
Short and long haul	REN3: Up to150 ft on 24 AWG line
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID

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Polarity Reversal / Wink	Yes
EMC	EN55022/EN55024 and FCC part15 Class B
Safety	UL

3.2 Hardware specification

The hardware specifications of the GL-VP-xxxx FXS series are detailed in Table 4.

TABLE 4: Hardware Specification Of GL-VP-62XX

Ports	2 or 4 FXS/FXO Ports
LAN interface	2 x RJ45 10/100Mbps (switched or routed)
LED	4 or 8 LEDs (GREEN)
Universal Switching Power Adaptor	Input: 100-240V AC, 50/60Hz, 0.5A Max Output: 9V DC, 2A UL certified
Dimension	225mm (L) x 135mm (W) x 35mm (H)
Weight	0.85 kg
Temperature	32~104°F / 0~40°C
Humidity	10% - 90% (non-condensing)
Compliance	FCC, CE

4 CALL FEATURES

The GL-VP-62XX supports the traditional telephony features available in a PBX as well as additional advanced telephony features.

TABLE 6: Call Features Table (Star Code)

Key	Call Features
***30**	**Block CallerID** (for all-config change)
***31**	**Send CallerID** (for all-config change)
***67**	**Block CallerID** (per call)
***82**	**Send CallerID** (per call)
***47**	**Direct IP Calling.** Dial “*47” + “IP address”. **No** dial tone will be

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	played in the middle. Detail see Direct IP Calling section on page 12.
***50**	**Disable Call Waiting** (for all-config change)
***51**	**Enable Call Waiting** (for all-config change)
***69**	**Call Return Service:** Dial *69 and the phone will dial the last incoming phone number received.
***70**	**Disable Call Waiting** (Per Call)
***71**	**Enable Call Waiting** (Per Call)
***72**	**Unconditional Call Forward:** Dial "*72" and then the forwarding number followed by "**#**". Wait for dial tone and hang up. (dial tone indicates successful forward)
***73**	**Cancel Unconditional Call Forward:** Dial "*73" and wait for dial tone, then hang up.
***78**	**Enable Do Not Disturb (DND):** When enabled all incoming calls will be rejected.
***79**	**Disable Do Not Disturb (DND):** When disabled, incoming calls will be accepted.
***87**	**Blind Transfer**
***90**	**Busy Call Forward:** Dial "*90" and then the forwarding number followed by "#". Wait for dial tone then hang up.
***91**	**Cancel Busy Call Forward**: dial "*91". Wait for dial tone. Hang up.
***92**	**Delayed Call Forward:** Dial "*92" and then the forwarding number followed by "**#**". Wait for dial tone then hang up.
***93**	**Cancel Delayed Call Forward:** Dial "*93" for a dial tone, then hang up.
Flash/Hook	If user hears call waiting beep, flash/hook will switch to the new incoming call. Also used to switch to a new channel for a new call.
#	Pressing pound sign will serve as **Re-Dial** key.

5 BASIC OPERATIONS

5.1 Understanding GL-VP-xxxx Voice Prompts

GL-VP-62XX has a stored voice prompt menu for quick browsing and simple configuration. To enter the voice prompt menu, press *** on the standard analog phone connected to any FXS port.

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TABLE 5: Definitions Of The GL-VP-xxxx Voice Prompts

Menu	Voice Will Say the Following:	
Main Menu	"Enter a Menu Option"	Enter "*" for the next menu option Enter "#" to return to the main menu Enter 01 – 05, 07,10 - 17, 47, 86 or 99 Menu option
01	"DHCP Mode", "PPPoE Mode" or "Static IP Mode"	Enter '9' to toggle the selection If user selects "Static IP Mode", user need configure all the IP address information through menu 02 to 05. If user selects "Dynamic IP Mode", the device will retrieve all IP address information from DHCP server automatically when user reboots the device.
02	"IP Address " + IP address	The current WAN IP address is announced Enter 12-digit new IP address if in Static IP Mode.
03	"Subnet " + IP address	Same as Menu option 02
04	"Gateway " + IP address	Same as Menu option 02
05	"DNS Server " + IP address	Same as Menu option 02
07	Preferred Vocoder	Enter "9" to go to the next selection in the list: ➢ PCM U ➢ PCM A ➢ iLBC ➢ G-726 ➢ G-723 ➢ G-729
10	"MAC Address"	Announces the Mac address of the unit.
12	WAN Port Web Access	Enter "9" to toggle between enable and disable
13	Firmware Server IP Address	Announces current Firmware Server IP address. Enter 12 digit new IP address.
14	Configuration Server IP Address	Announces current Config Server Path IP address. Enter 12 digit new IP address.
15	Upgrade Protocol	Upgrade protocol for firmware and configuration update. Enter "9" to toggle between TFTP and

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		HTTP
16	Firmware Version	Firmware version information.
17	Firmware Upgrade	Firmware upgrade mode. Enter “9” to rotate among the following three options: 1. always check 2. check when pre/suffix changes 3. never upgrade
47	“Direct IP Calling”	Enter the target IP address to make a direct IP call, after dial tone. (See “Make a Direct IP Call”.)
	“Invalid Entry”	Automatically returns to Main Menu

Five Success Tips when using the Voice Prompt

1. “*” shifts down to the next menu option

2. “**#**” returns to the main menu

3. “9” functions as the ENTER key in many cases to confirm an option

4. All entered digit sequences have known lengths - 2 digits for menu option and 12 digits for IP address. For IP address, add 0 before the digits if the digits are less than 3 (i.e. - 192.168.0.26 should be key in like 192168000026. No decimal is needed).

5. Key entry can not be deleted but the phone may prompt error once it is detected

5.2 Placing A Phone Call

5.2.1 Phone or Extension Numbers

1. Dial the number directly and wait for 4 seconds (Default “No Key Entry Timeout”); or

2. Dial the number directly and press # (Use **#** as dial key” must be configured in web configuration).

3. If you want to call another FXS port press ***7 and the number of port.

Examples:

1. Dial an extension directly on the same proxy, and then press the # or wait for 4 seconds.

2. Dial an outside number, first enter the prefix number (usually 1+ or international code) followed by the phone number. Press # or wait for 4 seconds. Check with your VoIP service provider for further details on prefix numbers.

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5.2.2 Direct IP Calls

Direct IP calling allows two parties, that is, a FXS Port with an analog phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy.

Elements necessary to completing a Direct IP Call:

1. Both GL-VP-xxxx and other VoIP Device, have public IP addresses, or
2. Both GL-VP-xxxx and other VoIP Device are on the same LAN using private IP addresses, or
3. Both GL-VP-xxxx and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

GL-VP-xxxx supports two ways to make Direct IP Calling:

Using IVR

1. Pick up the analog phone then access the voice menu prompt by dial "***"
2. Dial "47" to access the direct IP call menu
3. Enter the IP address using format ex. 192*168*0*160 after the dial tone.

Using Star Code

1. Pick up the analog phone then dial "*47"
2. Enter the target IP address using same format as above.

Note: NO dial tone will be played between step 1 and 2.

Destination ports can be specified by using "*" (encoding for ":") followed by the port number.

Examples:

a) If the target IP address is 192.168.0.160, the dialing convention is ***47 or Voice Prompt with option 47, then 192*168*0*160.** Followed by pressing the "#" key if it is configured as a send key or wait 4 seconds. In this case,
the default destination port 5060 is used if no port is specified.

b) If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be: ***47 or Voice Prompt with option 47, then 192*168*0*160*5062** followed by pressing the "#" key
If it is configured as a send key or wait for 4 seconds.

NOTE: When completing direct IP call, the "**Use Random Port**" should set to "NO". You cannot make direct IP calls between FXS1 to FXS2 since they are using same IP.

5.3 Call Hold

Place a call on hold by pressing the "flash" button on the analog phone (if the phone has that button).Press the "flash" button again to release the previously held Caller and resume conversation. If no "flash" button is available, use "hook flash" (toggle on-off hook quickly). You may drop a call using hook flash.

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5.4 Call Waiting

Call waiting tone (3 short beeps) indicates an incoming call, if the call waiting feature is enabled. Toggle between incoming call and current call by pressing the “flash” button. First call is placed on hold. Press the “flash” button to toggle between two active calls.

5.5 Call Transfer

Blind Transfer

Assume that call Caller A and B are in conversation. A wants to Blind Transfer B to C:

3. Caller A presses **FLASH** on the analog phone to hear the dial tone.

4. Caller A dials ***87** then dials caller C’s number, and then **#** (or wait for 4 seconds)

5. Caller A will hear the confirm tone. Then, A can hang up.

NOTE: “Enable Call Feature” must be set to “Yes” in web configuration page.

Caller A can place a call on hold and wait for one of three situations:

1. A quick confirmation tone (similar to call waiting tone) followed by a dial tone. This indicates the transfer is successful (transferee has received a 200 OK from transfer target). At this point, Caller A can either hang up or make another call.

2. A quick busy tone followed by a restored call (on supported platforms only). This means the transferee has received a 4xx response for the INVITE and we will try to recover the call. The busy tone is just to indicate to the transferor that the transfer has failed.

3. Continuous busy tone. The phone has timed out. Note: continuous busy tone does not indicate the transfer has been successful, nor does it indicate the transfer has failed. It often means there was a failure to receive second NOTIFY – check firmware for most recent release.

Attended Transfer

Assume that Caller A and B are in conversation. Caller A wants to Attend Transfer B to C:

1. Caller A presses **FLASH** on the analog phone for dial tone.

2. Caller A then dials Caller C’s number followed by **#** (or wait for 4 seconds).

3. If Caller C answers the call, Caller A and Caller C are in conversation. Then A can hang up to complete transfer.

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4. If Caller C does not answer the call, Caller A can press “flash” to resume call with Caller B.

NOTE: When Attended Transfer fails and A hangs up, the GL-VP-62XX will ring back user A to remind A that B is still on the call. A can pick up the phone to resume conversation with B.

5.6 3-Way Conferencing

The GL-VP-62XX supports Bellcore style 3-way Conference.

Instructions for 3-way conference:

Assuming that call party A and B are in conversation. A (GL-VP-62XX) wants to bring C in a conference:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.

2. A dials *23+C’s number then # (or wait for 4 seconds).

3. If C answers the call, then A presses FLASH to bring B, C in the conference.

4. If C does not answer the call, A can press FLASH back to talk to B.

5. Conference end after A hangs up.

5.7 PSTN Pass Through/Life Line

The RJ-11 line jack on the GL-VP-62XX functions as a pass through jack when the GL-VP-62XX is out of power. The pass through/life line enables the user to use the analog phone for PSTN calls directly without using an access code. Users will be able to use the same analog phone for PSTN calls.

To make a call using PSTN line press *00 at first.

5.8 Sending And Receiving Fax

GL-VP-62XX supports fax in two modes:

1) Fax Pass through. If the service provider does not support T.38, pass-through mode may be used. If you have problems with sending or receiving Fax, toggle the Fax Tone Detection Mode setting.

2) T.38 (Fax over IP)

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6 CONFIGURATION GUIDE

6.1 Configuration With Web Browser

The GL-VP-62XX has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow a user to configure the IP phone through any common web browser.

1. Connect the Power to the GL-VP-62XX unit.

2. Connect an Ethernet cable between the WAN port on GL-VP-62XX to your PC.

3. You will have to assign a dummy IP with the same subnet as the GL-VP-xxxx IP Address, which is 192.168.0.160 by default. So, set an IP address like 192.168.0.x for your PC.

4. Launch web browser and type http://192.168.0.160 at address of web browser. This connects you to the GL-VP-62XX web server.

You may choose to use DHCP or PPPoE connection or another static IP address according to your local network environment.

The Gateway Web Configuration Menu can be then accessed by the following URI: http://Gateway-IP-Address where the Gateway-IP-Address is the IP address of the Gateway.

NOTE: To access the configuration page, type the GL-VP-xxxx IP address into the browser, stripping out the leading “0” because the browser will parse in octet. e.g. if the IP address is: 192.168.001.014, please type in: 192.168.1.14.

6.2 End User Configuration

Once the HTTP request is entered and sent from a Web browser, the user will see a log in screen. There are two default passwords for the login page:

User	Password:	Level:
End User Level	1234	Only Status and Basic Settings
Administrator Level	admin	Browse all pages

After login, the next configuration page is the Basic Configuration page, explained in detail in Table 5 : Web Log-in Definition.

TABLE 5: Web Log-In Definitions (Basic Settings Page)

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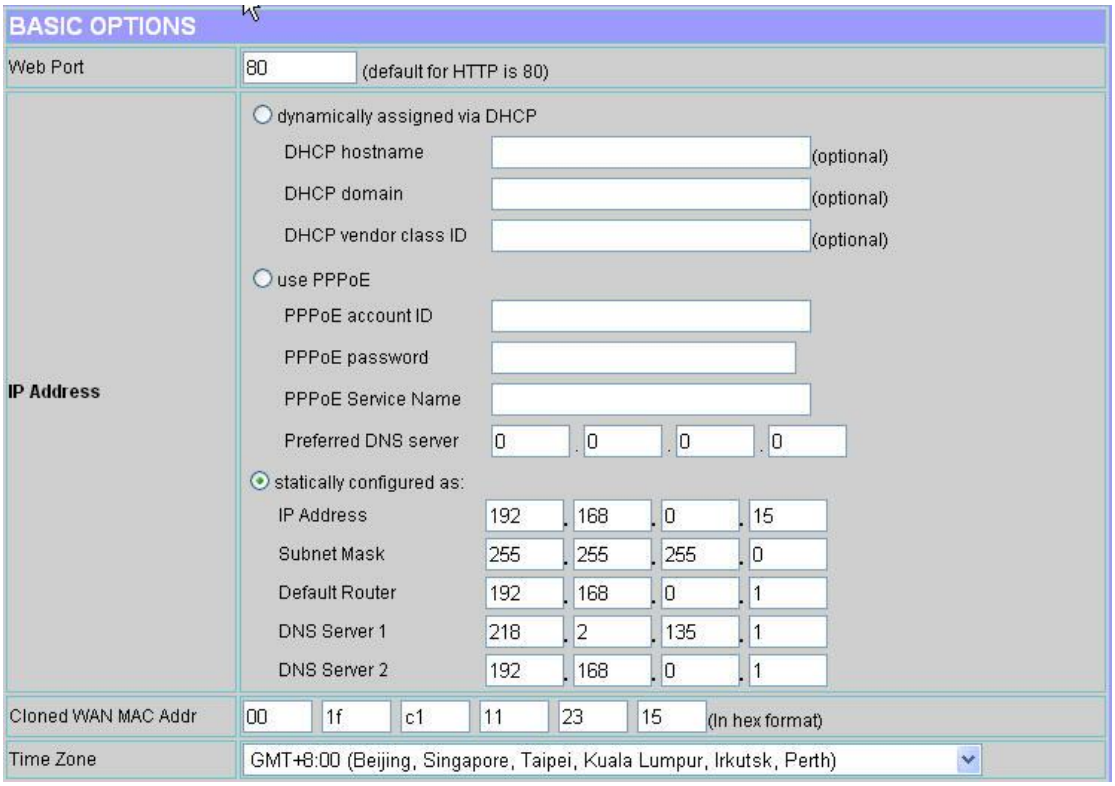

WEB LOG-IN DEFINITIONS (BASIC SETTINGS PAGE)	
Setting Options	**Meaning**
Web Port	By default, HTTP uses port 80. This field is for customizable web port.
IP Address	There are two modes to operate the GL-VP-62XX: **DHCP mode**: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The GL-VP-62XX acquires its IP address from the first DHCP server it discovers from the LAN it is connected. *Using the PPPoE feature*: set the PPPoE account settings. The GL-VP-62XX will establish a PPPoE session if any of the PPPoE fields is set. **Static IP mode:** configure the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields.
Time Zone	Controls how the date/time is displayed according to the specified time zone.

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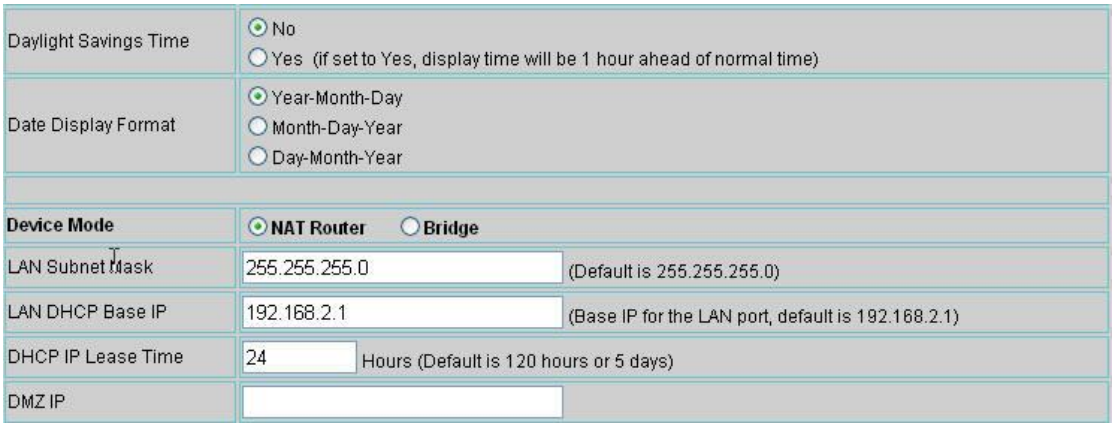

WEB LOG-IN DEFINITIONS (BASIC SETTINGS PAGE)	
Setting Options	**Meaning**
Daylight Savings Time	This parameter controls whether the displayed time will be daylight savings time or not. If set to Yes, then the displayed time will be 1 hour ahead of normal time.
Date Display Format	Allow user to choose among the following three formats: Year-Month-Day Month-Day-Year Day-Month-Year
Device Mode	This parameter controls whether the device is working in NAT router mode or Bridge mode. Save the setting and reboot prior to configuring the GL-VP-xxxx.
LAN Subnet Mask	Sets the LAN subnet mask. Default value is 255.255.255.0
LAN DHCP Base IP	Base IP for the LAN port which functions as a Gateway for the subnet. Default value is **192.168.22.1**.
DHCP IP Lease Time	Value is set in units of hours. Default value is **120 hrs** (5 Days.) The time IP address is assigned to the LAN clients.
DMZ IP	Forward all WAN IP traffic to a specific IP address if no matching port is used by GL-VP-62XX or defined in port forwarding.

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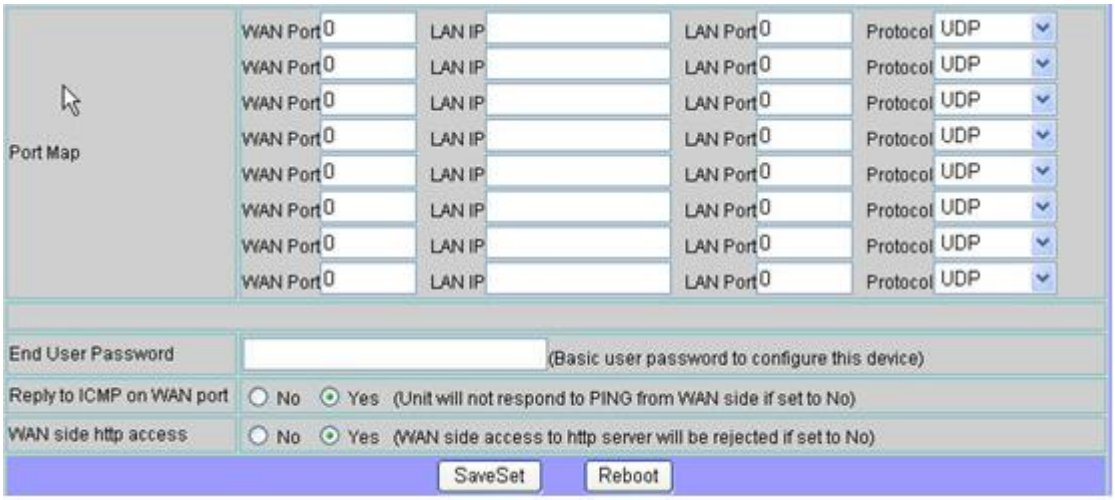

WEB LOG-IN DEFINITIONS (BASIC SETTINGS PAGE)	
Setting Options	**Meaning**
Port Map	Forwards a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port
End User Password	This contains the password to access the Web Configuration Menu. This field is case sensitive.
Reply to ICMP on WAN port	If set to “Yes”, the GL-VP-62XX will respond to the PING command from other computers, but it also is vulnerable to the DOS attack. Default is **No**.
WAN side http access	If this parameter is set to “No”, the HTML configuration update via WAN port is disabled.

You may also access the Device Status page which provides details of the GL-VP-xxxx product. The Device Status page terms are defined in Table 6: **Status Page Definitions.**

TABLE 6: Status Page Definitions

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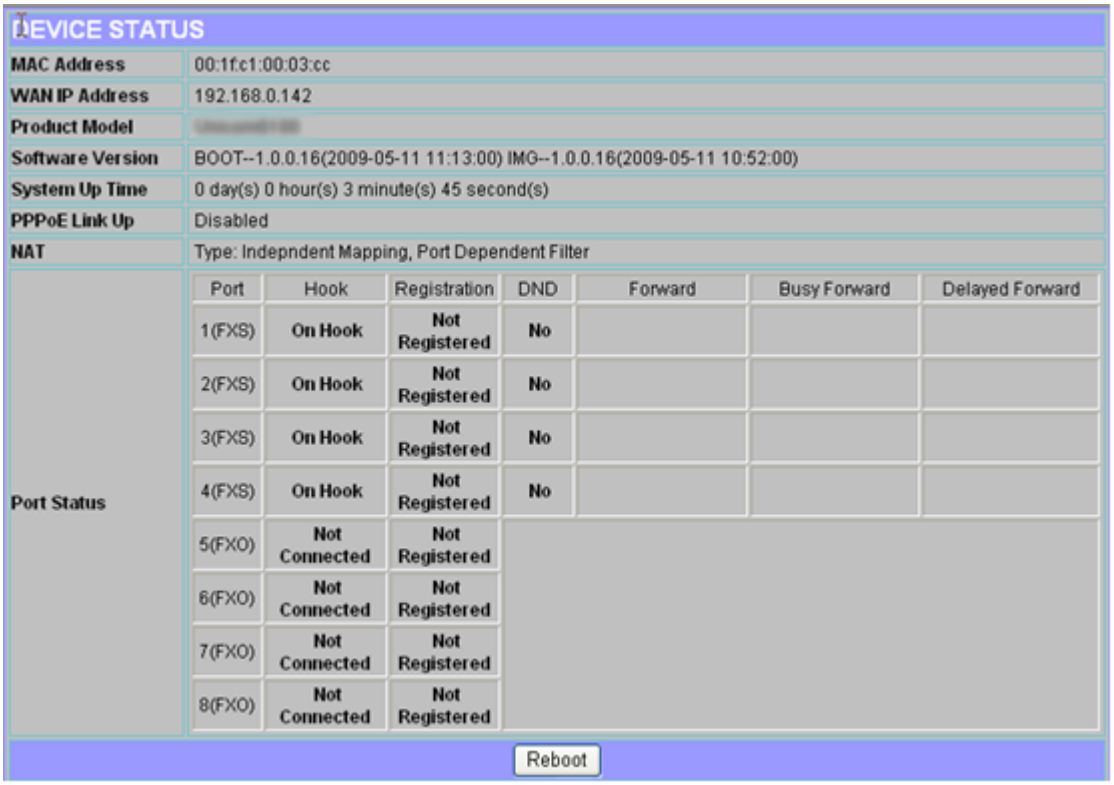

DEVICE STATUS

MAC Address	00:1f:c1:00:03:cc
WAN IP Address	192.168.0.142
Product Model	[redacted]
Software Version	BOOT--1.0.0.16(2009-05-11 11:13:00) IMG--1.0.0.16(2009-05-11 10:52:00)
System Up Time	0 day(s) 0 hour(s) 3 minute(s) 45 second(s)
PPPoE Link Up	Disabled
NAT	Type: Indepndent Mapping, Port Dependent Filter

Port Status

Port	Hook	Registration	DND	Forward	Busy Forward	Delayed Forward
1(FXS)	On Hook	Not Registered	No			
2(FXS)	On Hook	Not Registered	No			
3(FXS)	On Hook	Not Registered	No			
4(FXS)	On Hook	Not Registered	No			
5(FXO)	Not Connected	Not Registered				
6(FXO)	Not Connected	Not Registered				
7(FXO)	Not Connected	Not Registered				
8(FXO)	Not Connected	Not Registered				

Reboot

STATUS PAGE DEFINITIONS	
Setting Options	**Meaning**
MAC Address	The device ID in HEX format. This is needed for ISP troubleshooting. Note there are separate MAC addresses for the WAN side and the LAN side.
WAN IP Address	Shows WAN IP address of GL-VP-62XX
Product Model	Contains the product model info.
Software Version	Program: This is the main software release. Boot and Loader are not changed often.
System Up Time	Shows system up time since the last reboot.
PPPoE Link Up	Shows whether the PPPoE connection is running if connected to DSL modem.
NAT	Shows type of NAT the GL-VP-62XX is connected to via its WAN port. It is based on STUN protocol.
Port Status	Shows several information regarding the individual FXO ports.

6.3 Super User Settings

The end-user needs to login to the Super user configuration page the same way as for the

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basic configuration page.

FIGURE 4: Screenshot Of Super User Configuration

Super User configuration includes the end user configuration and Super configurations including: SIP configuration, Codec selection, NAT Traversal Setting and other miscellaneous configuration.

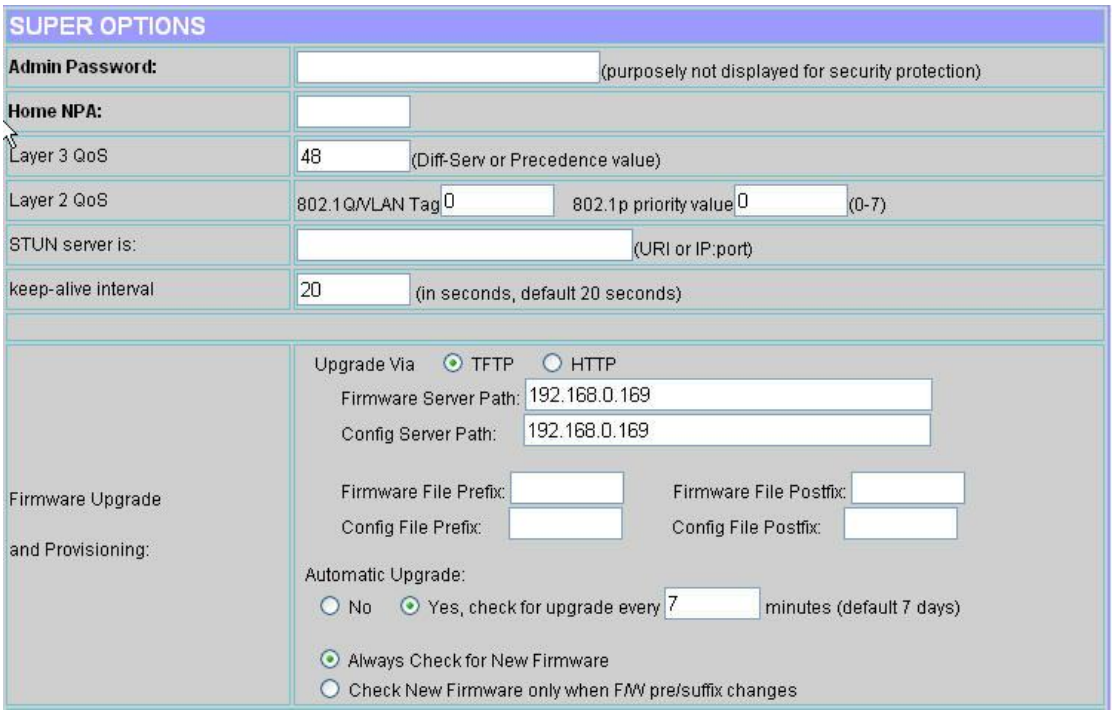

TABLE 7: Super Configuration Page Definitions

SUPER CONFIGURATION PAGE DEFINITIONS	
Setting Options	**Meaning**
Admin Password:	This contains the password to access the Super Web Configuration page.This field is case sensitive.
Home NPA:	Local area code for North American Dial Plan.
Layer 3 QoS	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is 48.
Layer 2 QoS	This contains the value used for layer 2 VLAN tag. Default setting is blank.
STUN server is:	IP address or domain name of stun server
keep-alive interval	This parameter specifies how often the GL-VP-62XX sends a blank UDP packet to the SIP server to keep the “hole” on the NAT open. Default is **20 seconds**.

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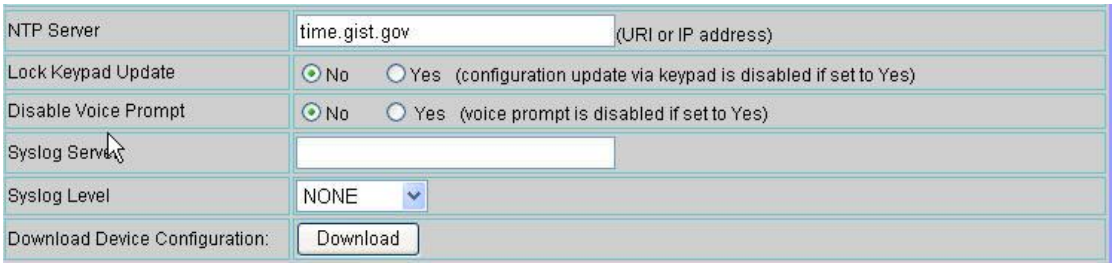
Firmware Upgrade and Provisioning:	Default method is HTTP. Firmware upgrade may take up to 10 minutes depending on network environment. Do not interrupt the firmware upgrading process.



SUPER CONFIGURATION PAGE DEFINITIONS	
Setting Options	**Meaning**
NTP Server	This parameter defines the URI or IP address of the NTP server which is used by the GL-VP-62XX to display the current date/time.
Lock Keypad Update	If this parameter is set to “Yes”, the configuration update via keypad is disabled.
Disable Voice Prompt	Default is No
Syslog Server	The IP address or URL of System log server. This feature is especially useful for the ITSP (Internet Telephone Service Provider)
Syslog Level	Select the GL-VP-62XX to report the log level. Default is NONE. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: 1. product model/version on boot up (INFO level) 2. NAT related info (INFO level) 3. sent or received SIP message (DEBUG level) 4. SIP message summary (INFO level) 5. inbound and outbound calls (INFO level) 6. registration status change (INFO level) 7. negotiated codec (INFO level) 8. Ethernet link up (INFO level) 9. SLIC chip exception (WARNING and ERROR levels) 10. memory exception (ERROR level)
Download Device Configuration	User can download configuration from the web page and save to configuration file.

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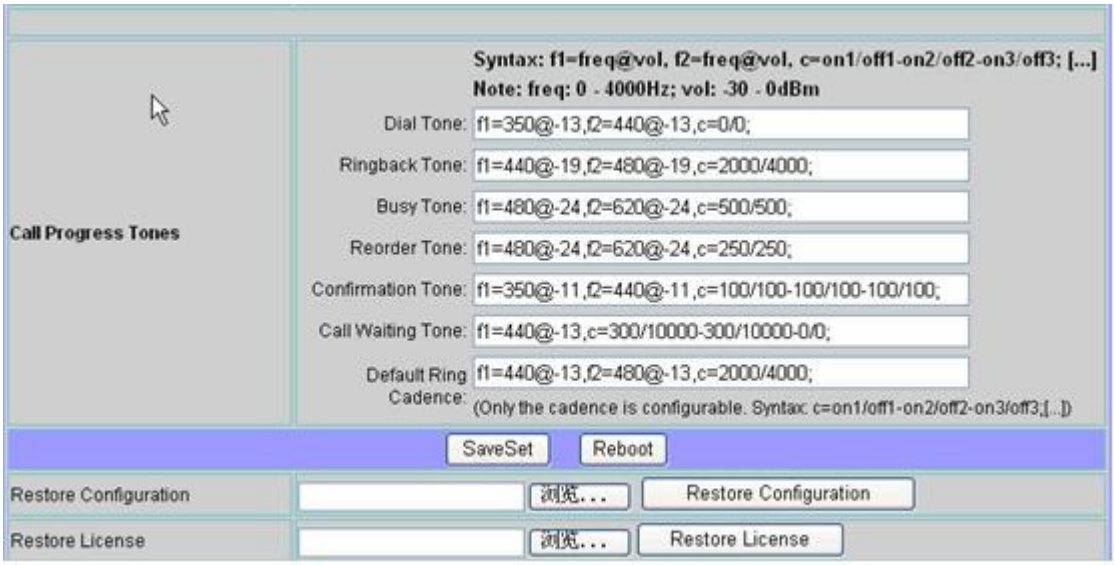

SUPER CONFIGURATION PAGE DEFINITIONS	
Setting Options	**Meaning**
Call Progress Tones	Using these settings, user can configure tone frequencies according to their preference. By default they are set to North American frequencies .Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. ON is the period of ringing ("On time" in 'ms') while OFF is the period of silence. In order to set a continuous tone, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern.
Restore Configuration	User can restore the before configuration from the configuration file saved at local pc

6.4 Configuring The FXS Channels

TABLE 10: FXS Ports Configuration Definitions

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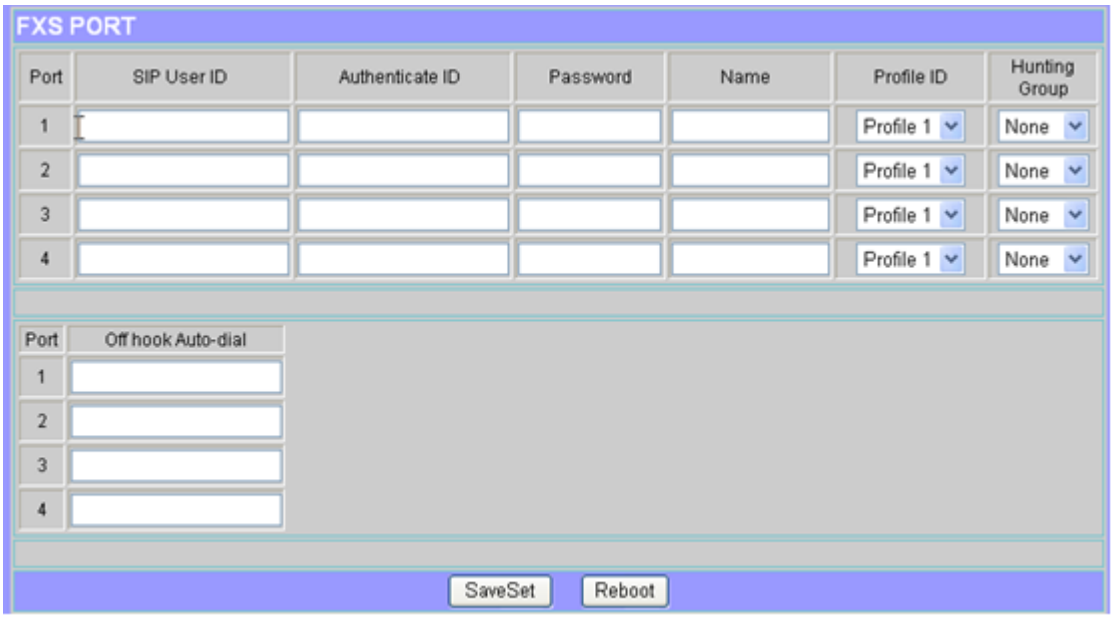

FXS PORT SETTING	
Setting Options	**Meaning**
FXS Port	FXS Port Number
SIP User ID	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
Password	SIP service subscriber's account password for GL-VP-62XX to register to (SIP) servers of ITSP.
Name	Name
Profile ID	Select the corresponding Profile ID (1/2)
Hunting Group	

6.5 FXS Profile

TABLE 11: FXS Profile Page Definitions

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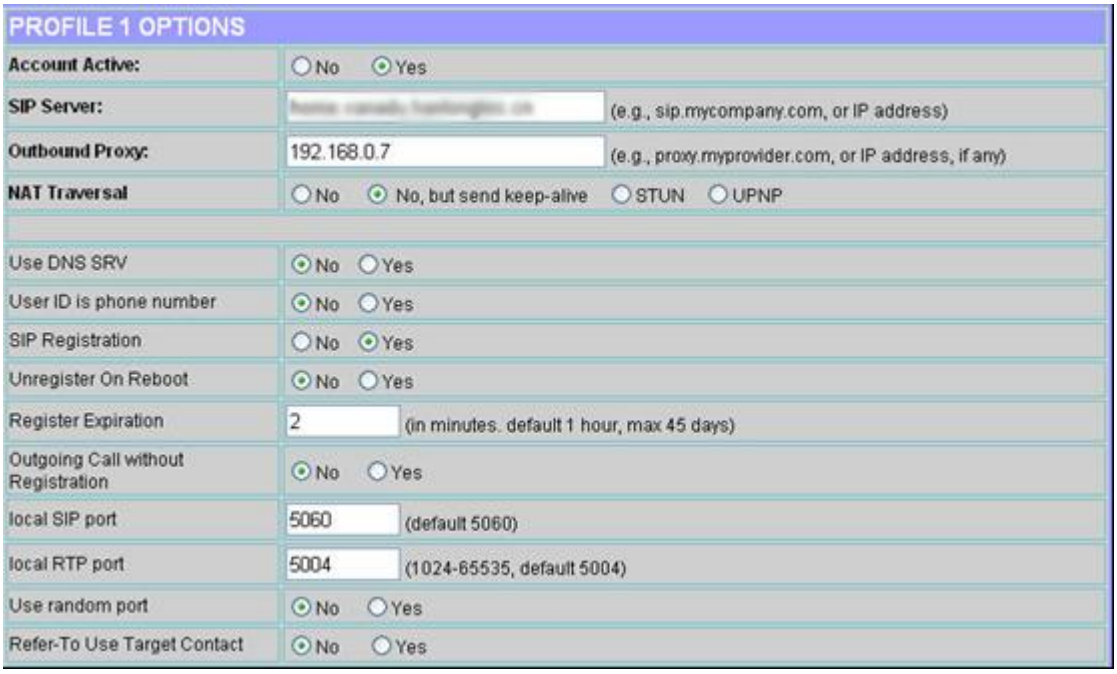
PROFILE 1 OPTIONS
Account Active: No Yes
SIP Server: (e.g., sip.mycompany.com, or IP address)
Outbound Proxy: 192.168.0.7 (e.g., proxy.myprovider.com, or IP address, if any)
NAT Traversal No No, but send keep-alive STUN UPNP
Use DNS SRV No Yes
User ID is phone number No Yes
SIP Registration No Yes
Unregister On Reboot No Yes
Register Expiration 2 (in minutes. default 1 hour, max 45 days)
Outgoing Call without Registration No Yes
local SIP port 5060 (default 5060)
local RTP port 5004 (1024-65535, default 5004)
Use random port No Yes
Refer-To Use Target Contact No Yes

PROFILE PAGE DEFINITIONS	
Setting Options	**Meaning**
Account Active	When set to Yes the SIP Profile is activated.
SIP Server	SIP Server's IP address or Domain name provided by VoIP service provider.
Outbound Proxy	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by GL-VP-62XX for firewall or NAT penetration in different network environments. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy can correct the problem.
NAT Traversal	This parameter defines whether the GL-VP-62XX NAT traversal mechanism is activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the GL-VP-62XX performs according to the STUN client specification. Under this mode, the embedded STUN client will detect if and what type of firewall/NAT is being used. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the GL-VP-62XX will use its mapped public IP address and port in all of its SIP and SDP messages. If the NAT Traversal field is set to "Yes" *with no specified STUN server*, the GL-VP-62XX will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the

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	NAT open. If your home or office router can act as a UPNP server, you can select “UPNP” option for NAT traversal.
Use DNS SRV:	Default is **No**. If set to “Yes” the client will use DNS SRV to look up server.
User ID is Phone Number	If the GL-VP-62XX has an assigned PSTN telephone number, this field should be set to “Yes”. Otherwise, set it to “No”. If “Yes” is set, a “user=phone” parameter will be attached to the “From” header in SIP request.
SIP Registration	This parameter controls whether the GL-VP-62XX needs to send REGISTER messages to the proxy server. The default setting is “Yes”.
Unregister on Reboot	Default is **No**. If set to “Yes”, the SIP user’s registration information is cleared on reboot.
Register Expiration	Allows the user to specify the time frequency (in minutes) for the GL-VP-62XX to refresh its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Outgoing Call without Registration	Default is **No**. If set to “Yes,” user can place outgoing calls even when not registered (if allowed by ITSP) but is unable to receive incoming calls.
Local SIP port	Defines the local SIP port the GL-VP-62XX will listen and transmit. The default value for Profile 1 is 5060 and 6060 for Profile 2.
Local RTP Port	Defines the local RTP-RTCP port pair the PROFILE will listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port _value for RTP and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP, port_value+3 for its RTCP and so on. The default value for Profile 1 is 5004 and 6004 for Profile 2.
Use random port	This parameter, when set to “YES”, will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple GL-VP-62XX are behind the same NAT.
Refer to Use Target Contact	Default is No. If set to Yes, then for Attended Transfer, the “Refer-To” header uses the transferred target’s Contact

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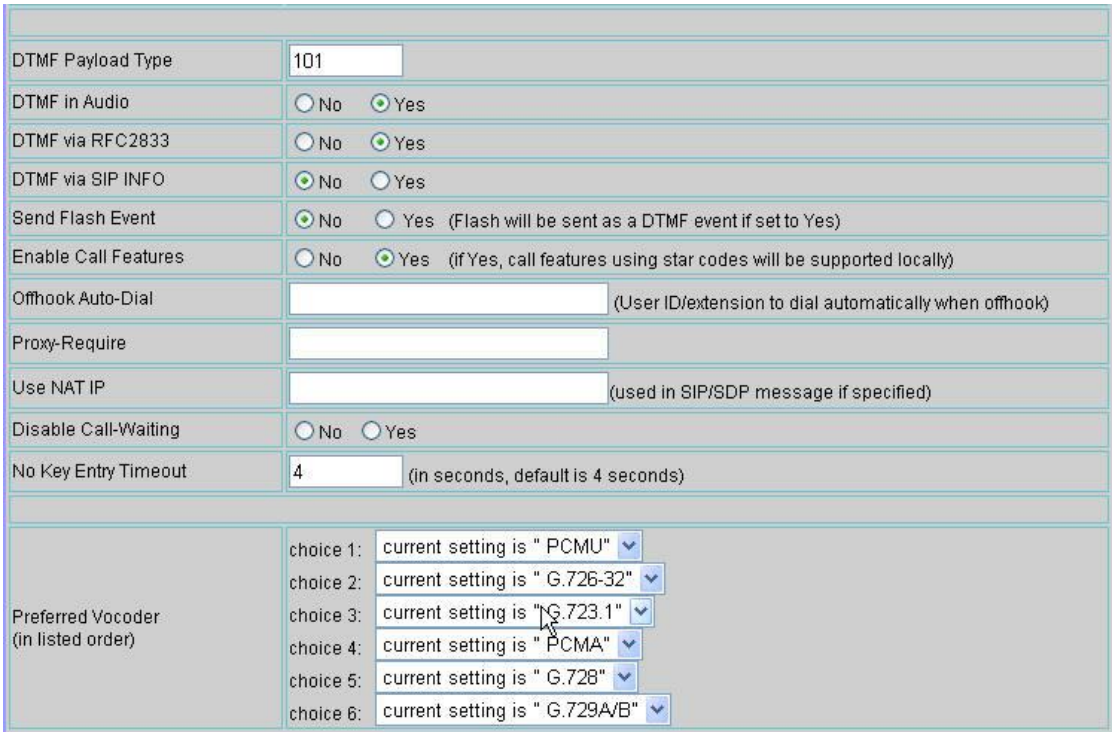
	header information.



PROFILE PAGE DEFINITIONS	
Setting Options	**Meaning**
DTMF Payload Type	Sets the payload type for DTMF using RFC2833.
DTMF in-audio	Send DTMF as inband (in-audio).
DTMF via RFC2833	Default "**YES**".
DTMF via SIP INFO	Send DTMF via SIP INFO message.
Send Flash Event	Default is **No**. If set to yes, flash will be sent as DTMF event.
Enable Call Features	Default is **Yes**. (If Yes, call features using star codes will be supported locally)
Off-Hook Auto Dial	Allows the user to configure a User ID or extension number to be automatically dialed upon off-hook. Only the user part of a SIP address needs to be entered here. The phone will automatically append the "@" and the host portion of the corresponding SIP address.
Proxy Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Use NAT IP	NAT IP address used in SIP/SDP message. Default is blank.
Disable Call Waiting	Default is **No**.

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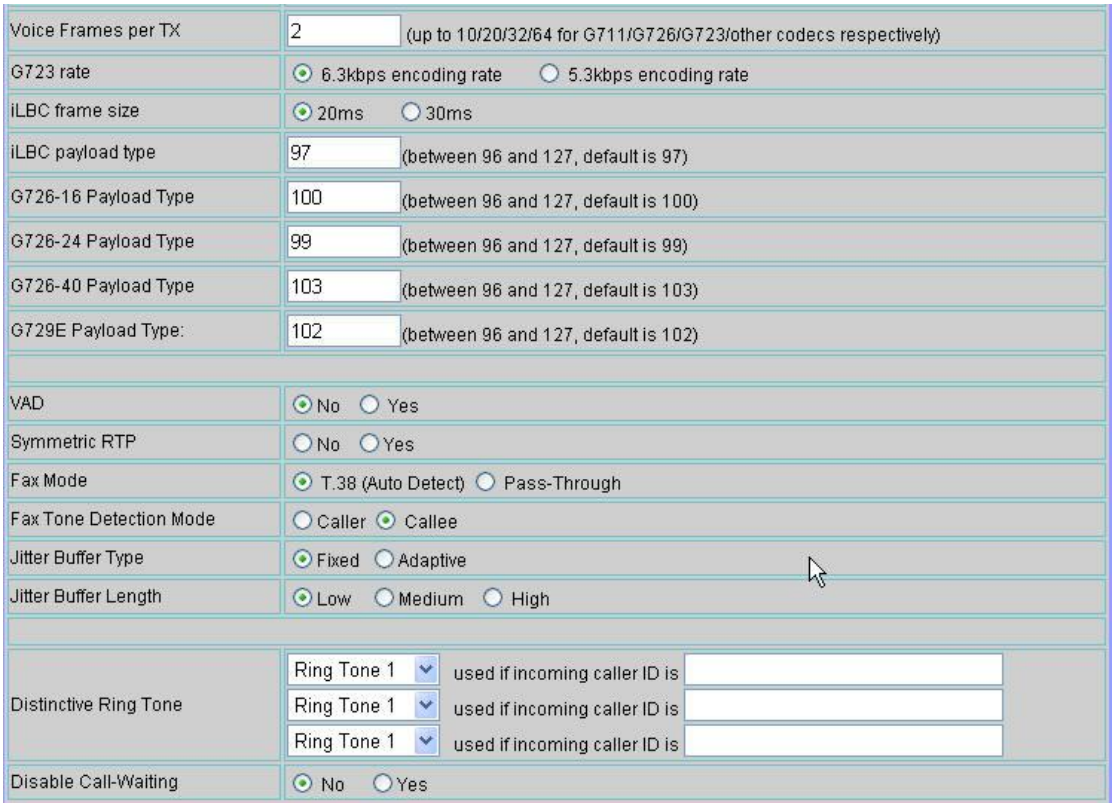
PROFILE PAGE DEFINITIONS

No Key Entry Timeout	Default is **4 seconds**.
Preferred Vocoder	The GL-VP-62XX supports up to 5 different Vocoder types including G.711 A-/U-law, G.726 (Supports bit rates 16, 24, 32 and 40), G.723.1, G.729A/B/E and iLBC. The user can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder is entered by choosing the appropriate option in “Choice 1”. The last Vocoder is entered by choosing the appropriate option in “Choice 8”.



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Setting Options	Meaning
Voice Frames per TX	This field contains the number of voice frames to be transmitted in a single packet. When setting this value, the user should be aware of the requested packet time (used in SDP message) as a result of configuring this parameter. This parameter is associated with the first vocoder in the above vocoder Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. e.g., if the first vocoder is configured as G723 and the "Voice Frames per TX" is set to be 2, then the "ptime" value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first vocoder chosen is G729 or G711 or G726, then the "ptime" value in the SDP message of an INVITE request will be 20ms. If the configured voice frames per TX exceeds the maximum allowed value, the GL-VP-62XX will use and save the maximum allowed value for the corresponding first vocoder choice. The maximum value for PCM is 10(x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames respectively.
G723 Rate	Defines the encoding rate for G.723 vocoder. By default, 6.3kbps rate is chosen.
iLBC Frame Size	Sets the iLBC frame size in 20ms or 30ms
iLBC Payload type	Defines payload type for iLBC. Default value is 97. The valid range is between 96 and 127.
G726-16 Payload type	Default value is 98. Range is from 96 to 127.
G726 – 24 Payload type	Default value is 99. Range is from 96 to 127.
G726 – 40 Payload type	Default value is 103. Range is from 96 to 127.
G729E payload type	Default value is 102. Range is from 96 to 127.
VAD	Default is **No**. VAD allows detecting the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network.
Symmetric RTP	Default is **No**. When set to Yes the device will change the destination to send RTP packets to the source IP address and port of the inbound RTP packet last received by the device.
Fax Mode	T.38 (Auto Detect) FoIP by default, or Pass-Through (must use codec PCMU/PCMA)

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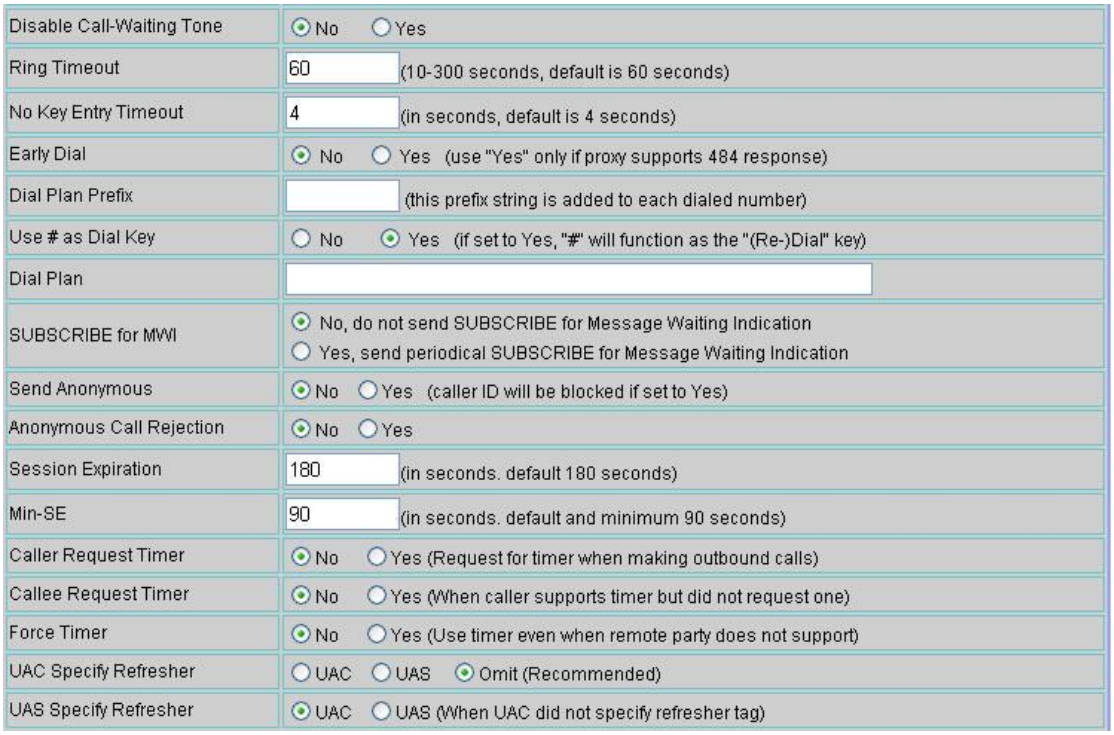
Fax Tone Detection Mode	Default is Callee. This decides whether Caller or Callee sends out the re-INVITE for T.38 or Fax Pass Through.
Jitter Buffer Type	Select either Fixed or Adaptive based on network conditions.
Jitter Buffer Length	Select Low, Medium or High based on network conditions.
Distinctive Ring tone	Custom Ring Tone 1 to 3 with associate Caller ID: when selected, if Caller ID is configured, then the device will ONLY uses this ring tone when the incoming call is from the Caller ID. System Ring Tone is used for all other calls. When selected but no Caller ID is configured, the selected ring tone will be used for **all** incoming calls.
Disable Call Waiting	Default is **No**.



PROFILE PAGE DEFINITIONS	
Setting Options	**Meaning**
Disable Call Waiting Tone	Default is **No**. This is to disable the stutter Call Waiting Tone when a Call Waiting call arrives. The CWCID will still be displayed.
Ring Timeout	Incoming call will stop ringing when not picked up given a specific period of time.
No Key Entry Timeout	Default is **4 seconds**.

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Early Dial	Default is **No**. Use only if proxy supports 484 response. This parameter controls whether the phone will send an early INVITE each time a key is pressed when a user dials a number. If set to "Yes", an INVITE is sent using the dial-number collected thus far; Otherwise, no INVITE is sent until the "(Re-)Dial" button is pressed or after about 5 seconds have elapsed if the user forgets to press the "Re-Dial" button. The "Yes" option should be used ONLY if there is a SIP proxy configured and the proxy server supports 484 Incomplete Address response. Otherwise, the call will likely be rejected by the proxy (with a 404 Not Found error). *This feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling*.
Dial Plan Prefix	Sets the prefix added to each dialed number.
Use # as Dial Key	Allows users to configure the "**#**" key as the "Send" (or "Dial") key. If set to "Yes", "**#**" will send the number. In this case, this key is essentially equivalent to the "Dial" key. If set to "No", this "#" key can be included as part of number.
Dial Plan	Dial Plan Rules: 1. Accept Digits: 1,2,3,4,5,6,7,8,9,0 , *, #, A,a,B,b,C,c,D,d 2. Grammar: x - any digit from 0-9; a. xx+ - at least 2 digits number; b. xx. ?at least 2 digits number; c. ^ - exclude; d. [3-5] - any digit of 3, 4, or 5; e. [147] - any digit 1, 4, or 7; f. <2=011> - replace digit 2 with 011 when dialing Example 1: {[369]11 | 1617xxxxxxx} Allow 311, 611, 911, and any 10 digit numbers of leading digits 1617 Example 2: {^1900x+ | <=1617>xxxxxxx} Block any number of leading digits 1900 and add prefix 1617 for any dialed 7 digit numbers Example 3: {1xxx[2-9]xxxxxx | <2=011>x+} Allow any length of number with leading digit 2 and 10 digit-numbers of leading digit 1 and leading exchange number between 2 and 9; if leading digit is 2, replace leading digit 2 with 011 before dialing. 3. Default: Outgoing - {x+} Example of a simple dial plan used in a Home/Office in the

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	US: { ^1900x. | <=1617>[2-9]xxxxxx | 1[2-9]xx[2-9]xxxxxx | 011[2-9]x. | [3469]11 } Explanation of example rule (reading from left to right): ^1900x. - prevents dialing any number started with 1900 <=1617>[2-9]xxxxxx - allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically 1[2-9]xx[2-9]xxxxxx |- allows dialing to any US/Canada Number with 11 digits length 011[2-9]x. - allows international calls starting with 011 [3469]11 - allow dialing special and emergency numbers 311, 411, 611 and 911 Note: In some cases user wishes to dial strings such as *123 to activate voice mail or other application provided by service provider. In this case * should be predefined inside dial plan feature and the Dial Plan should be: { [x*]+ }.
Subscribe for MWI	Default is **No**. When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Send Anonymous	If this parameter is set to "Yes", the "From" header along with Privacy and P_Asserted_Identity headers in outgoing INVITE message will be set to anonymous, blocking Caller ID.
Anonymous Call Rejection	Default is **No**. If set to Yes, incoming calls with anonymous Caller ID will be rejected with 610x Busy message.
Session Expiration	Default is 180 seconds.
Min-SE	Default is 90 seconds
Caller Request Timer	Default is NO
Callee Request Timer	Default is NO
Force Timer	Default is NO
UAC Specify Refresher	Default is Omit
UAS Specify Refresher	Default is UAC

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Force INVITE	◉ No ○ Yes (Always refresh with INVITE instead of UPDATE)
Special Feature	Standard
FXS Impedance	600 Ohm (North America)
Caller ID Scheme	Bellcore (North America)
Onhook Voltage	48V
Polarity Reversal	◉ No ○ Yes (reverse polarity upon call establishment and termination)
Hook Flash Timing	minimum: 200 maximum: 600 (Note: In 50-1200 milliseconds range)
Volume Amplification	TX 0dB default RX 0dB default

Ring Tones	**Syntax: c=on1/off1-on2/off2-on3/off3; [...]** Ring Tone 1 c=2000/4000; Ring Tone 2 c=2000/4000; Ring Tone 3 c=2000/4000; Ring Tone 4 c=2000/4000; Ring Tone 5 c=2000/4000; Ring Tone 6 c=2000/4000; Ring Tone 7 c=2000/4000; Ring Tone 8 c=2000/4000; Ring Tone 9 c=2000/4000; Ring Tone 10 c=2000/4000;
	SaveSet Reboot

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PROFILE PAGE DEFINITIONS	
Setting Options	**Meaning**
Force INVITE	Default is NO
Special Feature	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Broadsoft, etc.
FXS Impedance	Selects the impedance of the analog telephone connected to the Phone port.
Caller ID Scheme	Select the Caller ID Scheme to suit the standard of different area. • Bellcore (North America) • ETSI-FSK (France, Germany, Norway, Taiwan, UK-CCA) • ETSI-DTMF (Finland, Sweden) • DTMF (Denmark)
Onhook Voltage	Select the onhook voltage to suit different area or PBX
Polarity Reversal	Select Polarity Reversal to adapt some call charge/billing system. Default is No.
Hook Flash Timing	Time period when the cradle is pressed (Hook Flash) to simulate FLASH. To prevent unwanted activation of the Flash/Hold and automatic phone ring-back, adjust this time value.
Volume Amplification	Handset volume adjustment. RX is for receiving volume, TX is for transmission volume. Default values are 0dB for both parameters. +6dB generates the highest volume and -6dB generates the lowest volume.
Ring Tones	This function lets you configure ring tone cadence preferences. User has 10 choices. The configuration, completed in Distinctive Ring Tones block in the same page, applies to ring tones cadences configured here.

6.6 Configuring The FXO Channels

Configuring the FXO channels on the GL-VP-62XX is an easy process. Follow the GUI interfaces. The Device Status page terms are defined in **Table 8: FXO Lines**

Configuration Definitions. An example of the Channel Dialing Configuration is shown in

Figure 6. Please note the default is always configured. The user has the option to change the default settings as described in the Table 8.

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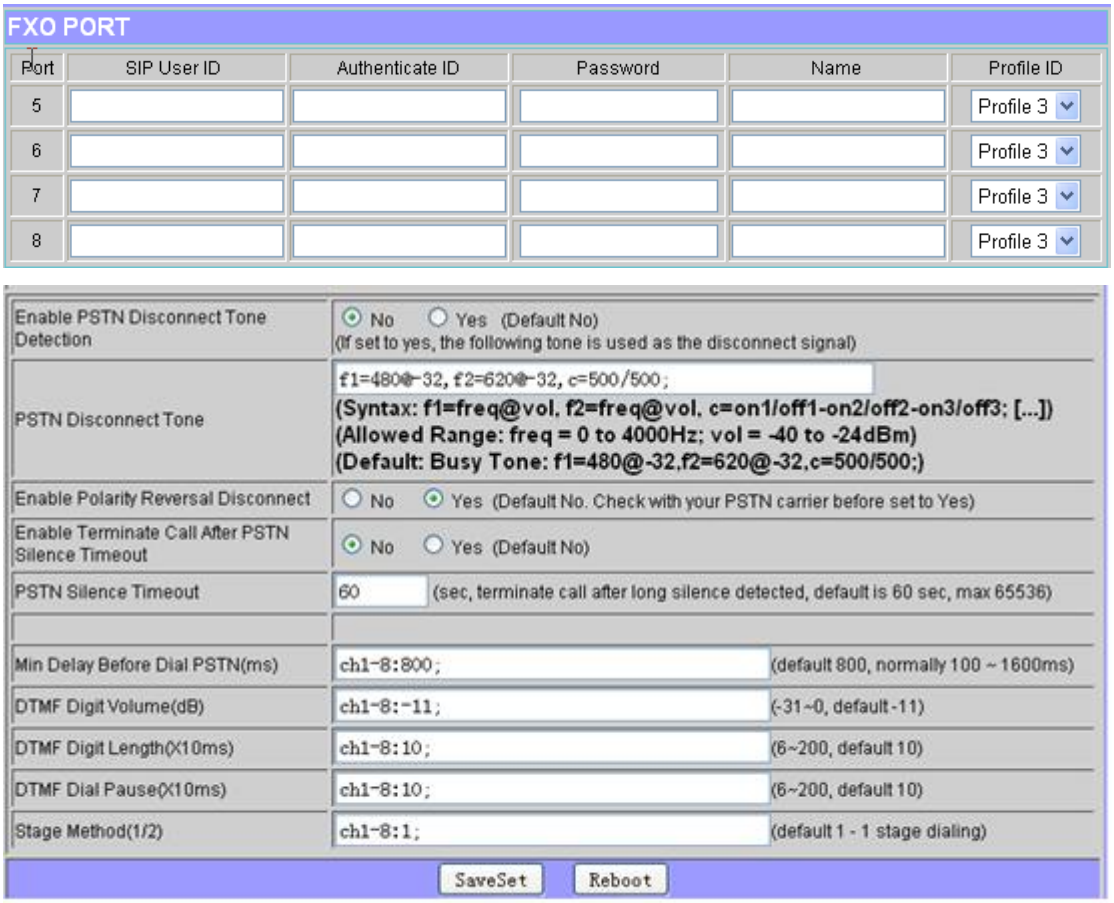

TABLE 8: FXO Lines Configuration Definitions

FXO PORT SETTING	
Setting Options	**Meaning**
FXO Port	FXO Port Number
SIP User ID	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
Password	SIP service subscriber's account password for GL-VP-62XX to register to (SIP) servers of ITSP.
Name	Name
Profile ID	Select the corresponding Profile ID (1/2)
Enable PSTN Disconnect Tone Detection	Default is No

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PSTN Disconnect Tone	Default: Busy Tone: f1=480@-32,f2=620@-32,c=500/500;)
Enable Polarity Reversal Disconnect	Default is No. This should be set to Yes only if the FXO lines are subscribed to PR service from PSTN Service provider. It is merely a PR detect feature. ***Note: If there is no PR service from provider on the FXO line, and this setting is configured to Yes, calls will not be successful.
Enable Terminate Call After PSTN Silence Timeout	Default is No
PSTN Silence Timeout	terminate call after long silence detected, default is 60 sec, max 65536
Min. Delay before Dial PSTN	Default is 800ms. This needs to be equal to or greater than the Current Disconnect threshold setting. Once the threshold is reached the gateway can dial out. This parameter should only be used if there are PSTN line detection issues.
DTMF Digit Volume(dB)	Default value is -11dB.
DTMF Digit Length(X10ms)	Default value is 100ms.
DTMF Dial Pause(X10ms)	Default value is 100ms.
Stage Method(1/2)	Syntax - ch1-8:1; {all channels 1 to 8 are set to value 1 or 2} Stage method can be set to either 1 or 2. Set this parameter to 1if you need to make a direct PSTN call from a VOIP endpoint. When you set it to 2, you will first dial one of the VOIP channel accounts from the VOIP endpoint, this will result in getting a PSTN line dial-tone to then dial out the destination PSTN number. Most implementations require this setting to be configured to 1.

6.7 FXO Profiles

Profiles are basically IP PBX / SIP Server configuration templates. If you have more than one IP PBX system or SIP Server that you would like to use with the GL-VP-62XX, then you can configure Profile 2 or 3. Note – Make sure you select the correct profile for each channel under Channels WEBPAGE.

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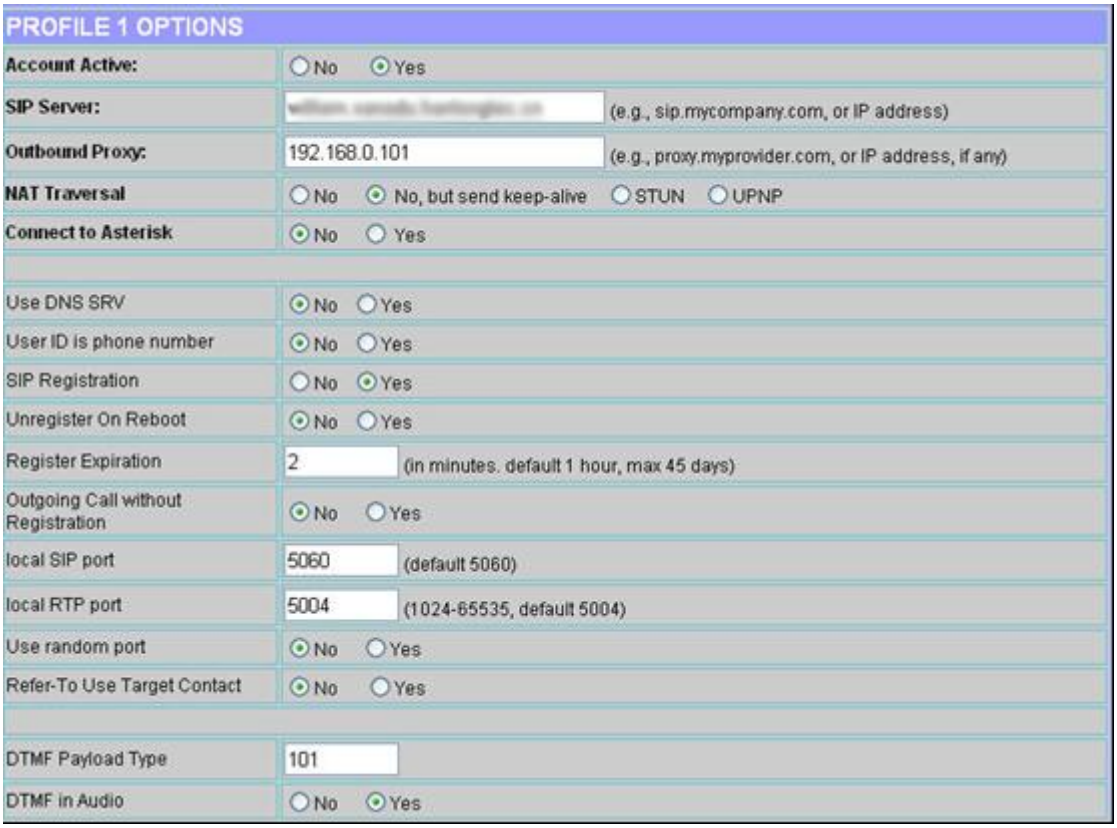
TABLE 9: Profile Page Definitions


PROFILE 1 OPTIONS
Account Active: No Yes
SIP Server: (e.g., sip.mycompany.com, or IP address)
Outbound Proxy: 192.168.0.101 (e.g., proxy.myprovider.com, or IP address, if any)
NAT Traversal No No, but send keep-alive STUN UPNP
Connect to Asterisk No Yes
Use DNS SRV No Yes
User ID is phone number No Yes
SIP Registration No Yes
Unregister On Reboot No Yes
Register Expiration 2 (in minutes. default 1 hour, max 45 days)
Outgoing Call without Registration No Yes
local SIP port 5060 (default 5060)
local RTP port 5004 (1024-65535, default 5004)
Use random port No Yes
Refer-To Use Target Contact No Yes
DTMF Payload Type 101
DTMF in Audio No Yes

PROFILE PAGE DEFINITIONS	
Setting Options	**Meaning**
Account Active:	When set to Yes the SIP Profile is activated.
SIP Server	SIP Server's IP address or Domain name provided by VoIP service provider.
Outbound Proxy	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by GL-VP-62XX for firewall or NAT penetration in different network environments. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy can correct the problem.
NAT Traversal	This parameter defines whether the GL-VP-62XX NAT traversal mechanism is activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the GL-VP-62XX performs according to the STUN client specification. Under this mode, the embedded STUN client will detect if and what type of firewall/NAT is being used. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the GL-VP-62XX will use its

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	mapped public IP address and port in all of its SIP and SDP messages. If the NAT Traversal field is set to "Yes" *with no specified STUN server*, the GL-VP-62XX will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.
Connect to Asterisk	If you want to register to Asterisk, please choose YES, If not, please choose NO
Use DNS SRV	Default is **No**. If set to "Yes" the client will use DNS SRV to look up server.
User ID is Phone Number	If the GL-VP-62XX has an assigned PSTN telephone number, this field should be set to "Yes". Otherwise, set it to "No". If "Yes" is set, a "user=phone" parameter will be attached to the "From" header in SIP request.
SIP Registration	This parameter controls whether the GL-VP-62XX needs to send REGISTER messages to the proxy server. The default setting is "Yes".
Unregister on Reboot	Default is **No**. If set to "Yes", the SIP user's registration information is cleared on reboot.
Register Expiration	Allows the user to specify the time frequency (in minutes) for the GL-VP-62XX to refresh its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Outgoing Call without Registration	Default is **No**. If set to "Yes," user can place outgoing calls even when not registered (if allowed by ITSP) but is unable to receive incoming calls.
Local SIP port	Defines the local SIP port the GL-VP-62XX will listen and transmit. The default value for Profile 1 is 5060 and 6060 for Profile 2.
Local RTP port	Defines the local RTP-RTCP port pair the PROFILE will listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port _value for RTP and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP, port_value+3 for its RTCP and so on. The default value for Profile 1 is 5004 and 6004 for Profile

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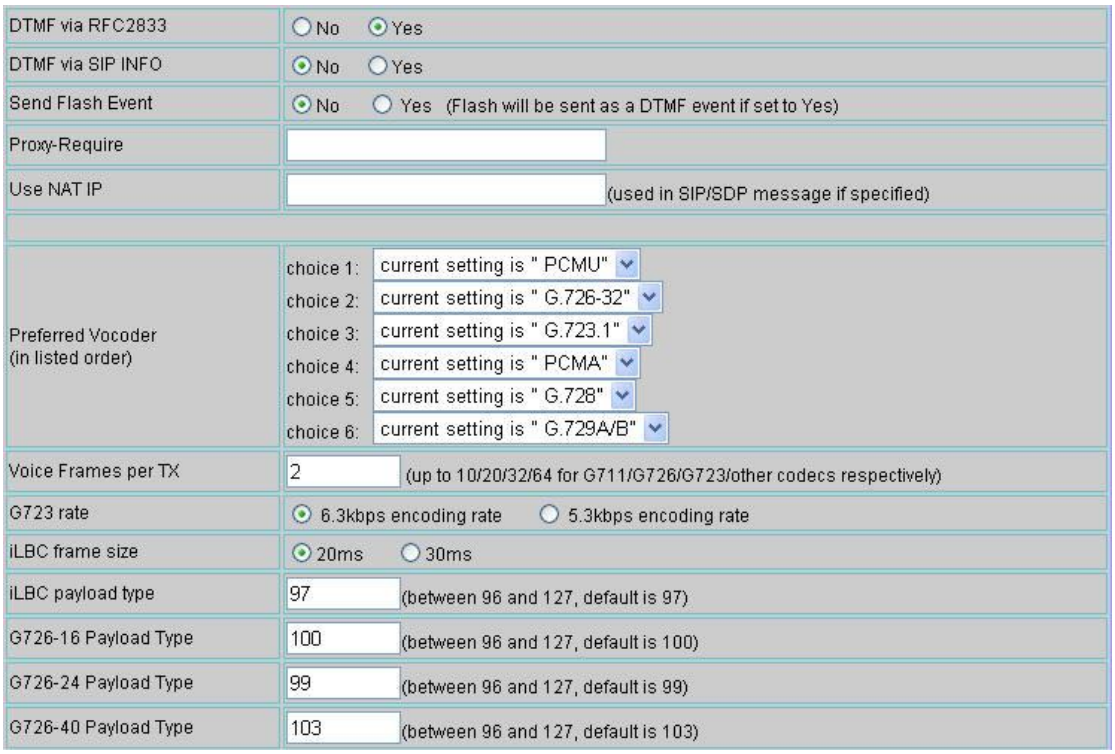
	2.
Use Random Port	This parameter, when set to “YES”, will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple GL-VP-62XX are behind the same NAT.
Refer-To Use Target Contact	Default is No. If set to Yes, then for Attended Transfer, the “Refer-To” header uses the transferred target’s Contact header information.
DTMF Payload Type	Sets the payload type for DTMF using RFC2833.
DTMF in Audio	Send DTMF as inband (in-audio).



PROFILE PAGE DEFINITIONS	
Setting Options	**Meaning**
DTMF via RFC2833	Default “**YES**”.
DTMF via SIP INFO	Send DTMF via SIP INFO message.
Send Flash Event	Default is **No**. If set to yes, flash will be sent as DTMF event.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Use NAT IP	NAT IP address used in SIP/SDP message. Default is blank.
Preferred Vocoder	The GL-VP-62XX supports up to 5 different Vocoder types

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(in listed order)	including G.711 A-/U-law, G.726 (Supports bit rates 16, 24, 32 and 40), G.723.1, G.729A/B/E and iLBC. The user can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder is entered by choosing the appropriate option in “Choice 1”. The last Vocoder is entered by choosing the appropriate option in “Choice 8”.
Voice Frames per TX	This field contains the number of voice frames to be transmitted in a single packet. When setting this value, the user should be aware of the requested packet time (used in SDP message) as a result of configuring this parameter. This parameter is associated with the first vocoder in the above vocoder Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. e.g., if the first vocoder is configured as G723 and the “Voice Frames per TX” is set to be 2, then the “ptime” value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first vocoder chosen is G729 or G711 or G726, then the “ptime” value in the SDP message of an INVITE request will be 20ms. If the configured voice frames per TX exceeds the maximum allowed value, the GL-VP-62XX will use and save the maximum allowed value for the corresponding first vocoder choice. The maximum value for PCM is 10(x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames respectively.
G723 Rate	Defines the encoding rate for G.723 vocoder. By default, 6.3kbps rate is chosen.
iLBC frame size	Sets the iLBC frame size in 20ms or 30ms
iLBC payload type	Defines payload type for iLBC. Default value is 97. The valid range is between 96 and 127.
G726-16 Payload Type	Default value is 97. Range is from 96 to 127.
G726-24 Payload Type	Default value is 99. Range is from 96 to 127.
G726-40 Payload Type	Default value is 103. Range is from 96 to 127.

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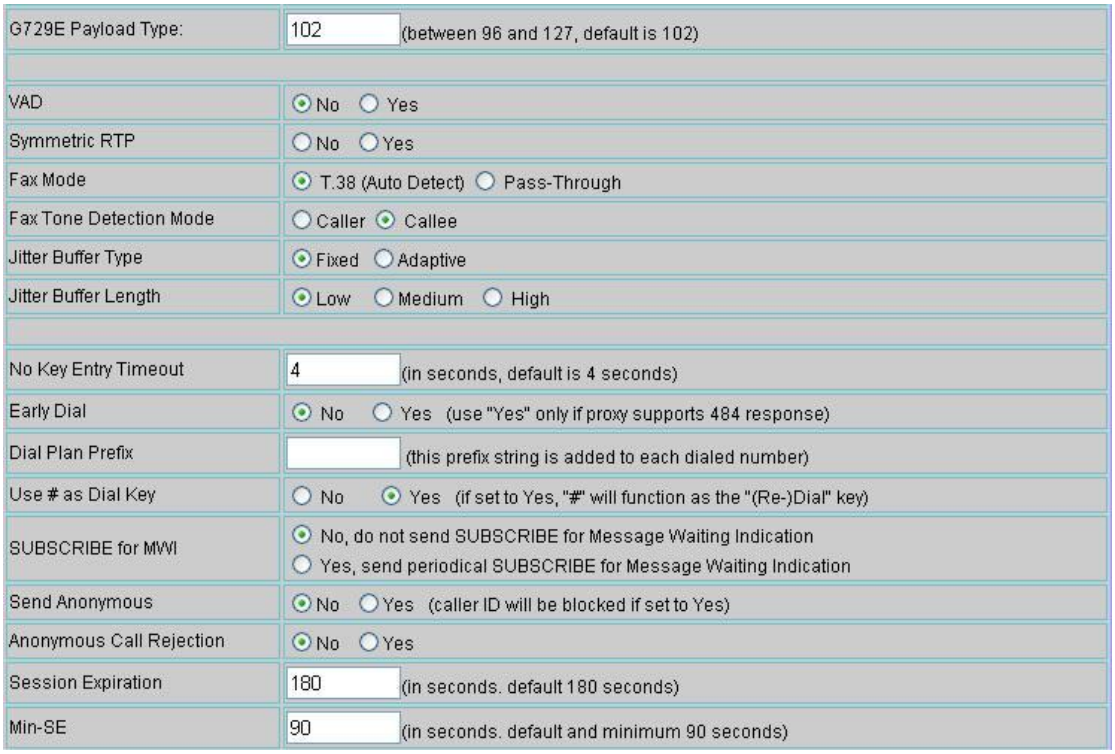

G729E Payload Type:	102 (between 96 and 127, default is 102)
VAD	No Yes
Symmetric RTP	No Yes
Fax Mode	T.38 (Auto Detect) Pass-Through
Fax Tone Detection Mode	Caller Callee
Jitter Buffer Type	Fixed Adaptive
Jitter Buffer Length	Low Medium High
No Key Entry Timeout	4 (in seconds, default is 4 seconds)
Early Dial	No Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix	(this prefix string is added to each dialed number)
Use # as Dial Key	No Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
SUBSCRIBE for MWI	No, do not send SUBSCRIBE for Message Waiting Indication Yes, send periodical SUBSCRIBE for Message Waiting Indication
Send Anonymous	No Yes (caller ID will be blocked if set to Yes)
Anonymous Call Rejection	No Yes
Session Expiration	180 (in seconds. default 180 seconds)
Min-SE	90 (in seconds. default and minimum 90 seconds)

PROFILE PAGE DEFINITIONS	
Setting Options	**Meaning**
G729E Payload Type:	Default value is 102. Range is from 96 to 127.
VAD	Default is **No**. VAD allows detecting the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network.
Symmetric RTP	Default is **No**. When set to Yes the device will change the destination to send RTP packets to the source IP address and port of the inbound RTP packet last received by the device.
Fax Mode	T.38 (Auto Detect) FoIP by default, or Pass-Through (must use codec PCMU/PCMA)
Fax Tone Detection Mode	Default is Callee. This decides whether Caller or Callee sends out the re-INVITE for T.38 or Fax Pass Through.
Jitter Buffer Type	Select either Fixed or Adaptive based on network conditions.
Jitter Buffer Length	Select Low, Medium or High based on network conditions.
No Key Entry Timeout	Default is **4 seconds**.
Early Dial	Default is **No**. Use only if proxy supports 484 response. This parameter controls whether the phone will send an early INVITE each time a key is pressed when a user dials a number.

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	If set to “Yes”, an INVITE is sent using the dial-number collected thus far; Otherwise, no INVITE is sent until the “(Re-)Dial” button is pressed or after about 5 seconds have elapsed if the user forgets to press the “Re-Dial” button. The “Yes” option should be used ONLY if there is a SIP proxy configured and the proxy server supports 484 Incomplete Address response. Otherwise, the call will likely be rejected by the proxy (with a 404 Not Found error). *This feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling*.
Dial Plan Prefix	Sets the prefix added to each dialed number.
Use # as Send Key	Allows users to configure the “**#**” key as the “Send” (or “Dial”) key. If set to “Yes”, “**#**” will send the number. In this case, this key is essentially equivalent to the “Dial” key. If set to “No”, this “#” key can be included as part of number.
Subscribe for MWI	Default is **No**. When set to “Yes” a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Send Anonymous	If this parameter is set to “Yes”, the “From” header along with Privacy and P_Asserted_Identity headers in outgoing INVITE message will be set to anonymous, blocking Caller ID.
Anonymous Call Rejection	Default is **No**. If set to Yes, incoming calls with anonymous Caller ID will be rejected with 610x Busy message.
Session Expiration	Default is 180 seconds.
Min-SE	Default is 90 seconds

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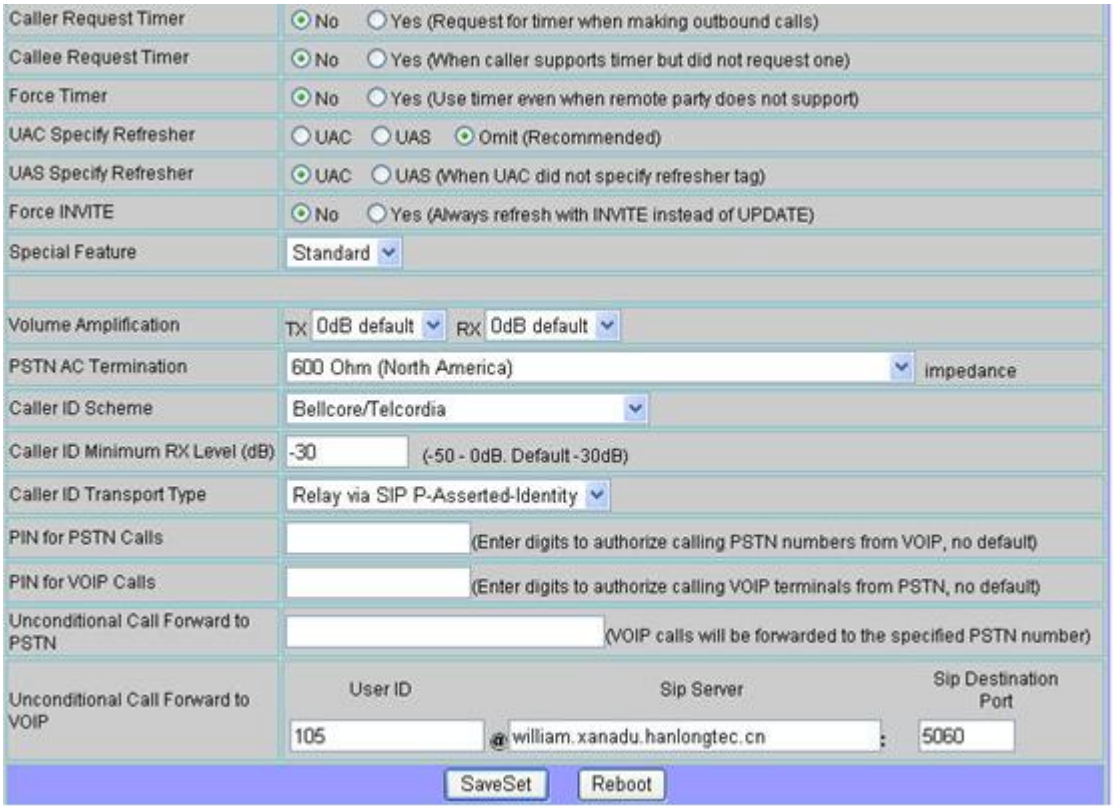

PROFILE PAGE DEFINITIONS	
Setting Options	**Meaning**
Caller Request Timer	Default is NO
Callee Request Timer	Default is NO
Force Timer	Default is NO
UAC Specify Refresher	Default is Omit
UAS Specify Refresher	Default is UAC
Force INVITE	Default is NO
Special Feature	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Broadsoft, etc.
Volume Amplification	Handset volume adjustment. RX is for receiving volume, TX is for transmission volume. Default values are 0dB for both parameters. +6dB generates the highest volume and -6dB generates the lowest volume.
PSTN AC Termination	Default is “600 Ohm(North America)”

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Caller ID Scheme	Select the Caller ID Scheme to suit the standard of different area. • Bellcore (North America) • ETSI-FSK (France, Germany, Norway, Taiwan, UK-CCA) • ETSI-DTMF (Finland, Sweden) • DTMF (Denmark)
Caller ID Minimum RX Level (dB)	Default is -30dB
Caller ID Transport Type	Default is “relay via From header”. You may also select :- “relay via P_Asserted_Identity header” “Disable” :- Caller ID feature will be disabled. “Send anonymous” :- All calls forwarded to VOIP end will be sent as anonymous.
PIN for PSTN Calls	Enter digits to authorize calling PSTN numbers from VOIP, no default
PIN for VOIP Calls	Enter digits to authorize calling VOIP terminals from PSTN, no default
Unconditional Call Forward to PSTN	VOIP calls will be forwarded to the specified PSTN number
Unconditional Call Forward to VOIP	This is an extremely important setting to make sure incoming PSTN calls are picked up and forwarded to the correct VOIP destination. User ID - This parameter allows users to configure a User ID or extension number to be automatically dialed upon FXO line off-hook. SIP Server - You also need to specify the Profile ofthe user id configured above (p1 stands for Profile 1, p2 stands for Profile 2 and so on). SIP Destination Port - Along with the user-id and Profile, you also have the option to choose the destination port where you would like to send the call. By default it should be set to ch1-x:5060; (x can be 4 or 8 depending on number of ports).

6.8 Saving The Configuration Changes

Once a change is made, press the “Update” button in the Configuration Menu. The GL-VP-62XX will display the following screen to confirm that the changes have been saved. To activate changes, reboot or power cycle the GL-VP-62XX after all changes are made.

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6.9 Rebooting From Remote

The administrator can remotely reboot the unit by pressing the “Reboot” button at the bottom of the configuration menu. The following screen will indicate that rebooting is underway.

The user can re-login to the unit after waiting for about 30 seconds.

7 SOFTWARE UPGRADE

To upgrade software, GL-VP-62XX can be configured with a TFTP server where the new code image is located. The TFTP upgrade can work in either static IP or DHCP mode using private or public IP address. It is recommended to set the TFTP server address in either a public IP address or on the same LAN with the GL-VP-62XX.

There are two ways to set up the TFTP server to upgrade the firmware, namely through voice menu prompt or via the GL-VP-62XX’s Web configuration interface. To configure the TFTP server via voice prompt, follow section 5.1 with option 06, once set up the TFTP IP address, power cycle the ATA, the firmware will be fetched once the ATA boots up.

To configure the TFTP server via the Web configuration interface, open up your browser to point at the IP address of the GL-VP-62XX. Input the admin password to enter the configuration screen. From there, enter the TFTP server address in the designated field towards the bottom of the configuration screen.

Once the TFTP server is configured, please power cycle the GL-VP-62XX.

TFTP process may take as long as 1 to 2 minutes over the Internet, or just 30+ seconds if it is performed on a LAN. Users are recommended to conduct TFTP upgrade in a controlled LAN environment if possible. For those who do not have a local TFTP server, Gigalink Co LTD provides a NAT-friendly TFTP server on the public Internet for firmware upgrade. Please check the Service section of Gigalink’s Web site to obtain this TFTP server’s IP address.

NOTES:

When Gigalink ATA boot up, it will send TFTP or HTTP request to download configuration files, there are two configuration files, one is “cfg.bin” and the other is “cfg001fc1xxxxxx”, where “001fc1xxxxxx” is the MAC address of the GL-VP-62XX. These two files are for initial automatically provisioning purpose only, for normal TFTP or HTTP firmware upgrade, the following error messages in a TFTP or HTTP server log can be ignored.

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8 RESTORE FACTORY DEFAULT SETTINGS

WARNING! Restoring the Factory Default Setting will DELETE all configuration information of the phone.

Please BACKUP or PRINT out all the settings before you approach to following steps. Gigalink will not take any responsibility if you lose all the parameters of setting and cannot connect to your VoIP service provider.

FACTORY RESET

There are two (2) methods for resetting your unit:

Reset Button

Reset default factory settings following these four (4) steps:

1. Unplug the Ethernet cable.

2. Locate a needle-sized hole on the back panel of the gateway unit next to the power connection.

3. Insert a pin in this hole, and press for about 8 seconds.

4. Take out the pin. All unit settings are restored to factory settings.

IVR Command

Reset default factory settings using the IVR Prompt (Table 5):

1. Dial “***” for voice prompt.

2. Enter “99” and wait for “reset” voice prompt.

3. Enter 862584658050

NOTE:

1. Factory Reset will be disabled if the **“Lock keypad update”** is set to “Yes”.

2. Please be aware by default the GL-VP-62XX WAN side HTTP access is disabled. After a factory reset, the device’s web configuration page can be accessed only from its LAN port.

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9 TECHNICAL SUPPORT CONTACT

Email: info@giga-link.ru

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